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EXAMINER	
ENSEY, BRIAN	
ART UNIT	PAPER NUMBER
2643	9

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Please find below and/or attached an Office communication concerning this application or proceeding.

Office Action Summary	Application No.	Applicant(s)
	10/023,264	NORDQVIST ET AL.
	Examiner	Art Unit
	Brian Ensey	2643

-- The MAILING DATE of this communication appears on the cover sheet with the correspondence address --

Period for Reply

A SHORTENED STATUTORY PERIOD FOR REPLY IS SET TO EXPIRE 3 MONTH(S) FROM THE MAILING DATE OF THIS COMMUNICATION.

- Extensions of time may be available under the provisions of 37 CFR 1.136(a). In no event, however, may a reply be timely filed after SIX (6) MONTHS from the mailing date of this communication.
- If the period for reply specified above is less than thirty (30) days, a reply within the statutory minimum of thirty (30) days will be considered timely.
- If NO period for reply is specified above, the maximum statutory period will apply and will expire SIX (6) MONTHS from the mailing date of this communication.
- Failure to reply within the set or extended period for reply will, by statute, cause the application to become ABANDONED (35 U.S.C. § 133). Any reply received by the Office later than three months after the mailing date of this communication, even if timely filed, may reduce any earned patent term adjustment. See 37 CFR 1.704(b).

Status

1)☒ Responsive to communication(s) filed on 4/26/04.

2a)☐ This action is **FINAL**. 2b)☒ This action is non-final.

3)☐ Since this application is in condition for allowance except for formal matters, prosecution as to the merits is closed in accordance with the practice under *Ex parte Quayle*, 1935 C.D. 11, 453 O.G. 213.

Disposition of Claims

4)☒ Claim(s) 1-19 and 21-27 is/are pending in the application.

4a) Of the above claim(s) _____ is/are withdrawn from consideration.

5)☐ Claim(s) _____ is/are allowed.

6)☒ Claim(s) 1-5,10,14,15,17 and 22-27 is/are rejected.

7)☒ Claim(s) 6-9,11-13,16,18,19 and 21 is/are objected to.

8)☐ Claim(s) _____ are subject to restriction and/or election requirement.

Application Papers

9)☐ The specification is objected to by the Examiner.

10)☐ The drawing(s) filed on _____ is/are: a)☐ accepted or b)☐ objected to by the Examiner.

Applicant may not request that any objection to the drawing(s) be held in abeyance. See 37 CFR 1.85(a).

Replacement drawing sheet(s) including the correction is required if the drawing(s) is objected to. See 37 CFR 1.121(d).

11)☐ The oath or declaration is objected to by the Examiner. Note the attached Office Action or form PTO-152.

Priority under 35 U.S.C. § 119

12)☐ Acknowledgment is made of a claim for foreign priority under 35 U.S.C. § 119(a)-(d) or (f).

a)☐ All b)☐ Some * c)☐ None of:

1.☐ Certified copies of the priority documents have been received.

2.☐ Certified copies of the priority documents have been received in Application No. _____.

3.☐ Copies of the certified copies of the priority documents have been received in this National Stage application from the International Bureau (PCT Rule 17.2(a)).

* See the attached detailed Office action for a list of the certified copies not received.

Attachment(s)

1) ☐ Notice of References Cited (PTO-892)
2) ☐ Notice of Draftsperson's Patent Drawing Review (PTO-948)
3) ☐ Information Disclosure Statement(s) (PTO-1449 or PTO/SB/08) Paper No(s)/Mail Date _____.
4) ☐ Interview Summary (PTO-413) Paper No(s)/Mail Date. _____ .
5) ☐ Notice of Informal Patent Application (PTO-152)
6) ☐ Other: _____.

DETAILED ACTION

Claim Rejections - 35 USC § 103

The text of those sections of Title 35, U.S. Code not included in this action can be found in a prior Office action.

Claims 1-5, 10, 14, 15, 17 and 22-27 are rejected under 35 U.S.C. 103(a) as being unpatentable over Allegro et al, U.S. Patent Application Publication US2002/0037087 A1 in view of Rahim, European Patent Application EP 0,881,625 A2.

Regarding claim 1, Allegro discloses a hearing aid prosthesis (1) comprising: a microphone (2) adapted to generate an input signal in response to receiving an acoustic signal from a listening environment, an output transducer (6) for converting a processed output signal into an electrical or an acoustic signal, processing means (4,5,7) adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal, a memory area (8) storing values of the related algorithm parameters for the predetermined signal processing algorithm, and processing the characteristics of the input signal with at least one discrete Hidden Markov Model, $\lambda^{source} = \{A^{source}, B^{source}, \alpha_0^{source}\}$, associated with a predetermined sound source to determine element value(s) of a classification vector indicating a probability of the predetermined sound source being active in the listening environment, control one or several values of the related algorithm parameters in dependence of the element value(s) of the classification vector; thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment; wherein; A_{source} = A state transition probability matrix; B_{source} = An observation

symbol probability distribution matrix for an input observation O(t) for each state of the at least one Hidden Markov Model (HMM); α_0^{source} = An initial state probability distribution vector (See Fig. 1 and paragraphs 0010-0027). Allegro does not expressly disclose segmenting the input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, O(t), representing predetermined signal features of the consecutive frames, wherein the feature vectors comprise a plurality of frequency-domain parameters or a plurality of time-domain parameters or any combination thereof, compare each of the feature vectors, O(t), with a feature vector set to determine, for substantially each feature vector, an associated symbol value so as to generate an observation sequence of symbol values associated with the consecutive signal frames. However, Allegro teaches extraction of characteristic features from an acoustic signal during an extraction phase (See paragraph 0019). Further, Rahim teaches a speech recognition system using a feature extractor to generate a set of feature vectors and a classification processor to generate a set of recognition models to be processed by one or more HMMs (See col. 6, line 26 to col. 7, line 35). Since speech occurs in the time-domain, it is inherent that the feature vectors comprise a plurality of time-domain parameters. Frequency-domain conversion is well-known in the art and is frequently used for efficient data processing. It would have been obvious to one of ordinary skill in the art at the time of the invention that input segmentation and vector generation is an integral part of processing an input signal using a Hidden Markov Model for determining the real time status of the acoustic scene.

Regarding claim 2, Allegro further discloses the processing means are adapted to process the input vectors with a plurality of discrete Hidden Markov Models associated with respective predetermined sound sources to determine the element values of the classification vector

indicating a probability of each predetermined sound source (See paragraphs 23-26). Allegro does not expressly disclose the creation of symbol values for associated feature vectors. However, Rahim teaches a speech recognition system using a feature extractor to generate a set of feature vectors and a classification processor to generate a set of recognition models to be processed by one or more HMMs (See col. 6, line 26 to col. 7, line 35). It would have been obvious to one of ordinary skill in the art at the time of the invention that input segmentation and vector generation is an integral part of processing an input signal using a Hidden Markov Model for determining the real time status of the acoustic scene.

Regarding claims 3 and 4, Allegro does not expressly disclose feature vectors are associated with respective integer symbol values during vector quantization process wherein the feature vector set comprises between 8 and 256 discrete symbols. However, Rahim teaches a speech recognition system using a feature extractor to generate a set of feature vectors and a classification processor to generate a set of recognition models to be processed (See col. 6, line 26 to col. 7, line 35). Rahim also teaches data is divided into N environments wherein N is represented by a positive integer. It would have been obvious to one of ordinary skill in the art at the time of the invention to use a positive integer set for the feature vectors such as a 0 and 1 for a binary system which uses a standard 8 bit memory to provide feature vector set comprises between 8 and 256 discrete symbols (2^8) since they are cheap and easily obtainable.

Regarding claim 5, Allegro further discloses the feature vector set has been determined in an off-line training procedure which utilized real-life sound source recordings and stored in non-volatile memory locations of the hearing instrument (See paragraphs 16, 25 and 26).

Regarding claim 10, Allegro discloses a hearing aid prosthesis (1) comprising: a microphone (2) adapted to generate an input signal in response to receiving an acoustic signal from a listening environment, an output transducer (6) for converting a processed output signal into an electrical or an acoustic signal, processing means (4,5,7)adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal, a memory area (8) storing values of the related algorithm parameters for the predetermined signal processing algorithm, and the processing means using a combination of HMMs and classifiers for multistage recognition processes to control one or several values of the related algorithm parameters in dependence of the element value(s) of the classification vector; thereby adapting characteristics of the predetermined signal processing algorithm to the current listening environment (See Fig. 1 and paragraphs 0010-0027). Allegro does not expressly disclose the processing means being further adapted to: segment the input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, O(t), representing predetermined signal features of the consecutive frames, wherein the feature vectors comprise a plurality of frequency-domain parameters or a plurality of time-domain parameters or any combination thereof, process the feature vectors with one or a plurality of Hidden Markov Models operating on a first time scale and associated with respective a predetermined sound source to determine element value(s) of a classification vector indicating a probability of the predetermined sound source being active in the listening environment, process the first classification vector with a Hidden Markov Model operating at a second time scale and associated with one or more predetermined sound sources to determine element value(s) of a classification vector, However, Allegro teaches extraction of

characteristic features from an acoustic signal during an extraction phase. Further, Rahim teaches a speech recognition system using a feature extractor to generate a set of feature vectors and a classification processor to generate a set of recognition models to be processed by one or more HMMs (See col. 6, line 26 to col. 7, line 35). Since speech occurs in the time-domain, it is inherent that the feature vectors comprise a plurality of time-domain parameters. Frequency-domain conversion is well-known in the art and is frequently used for efficient data processing. It would have been obvious to one of ordinary skill in the art at the time of the invention that input segmentation and vector generation is an integral part of processing an input signal using multiple Hidden Markov Models for determining the real time status of the acoustic scene.

Regarding claim 14, Allegro discloses a prosthesis as claimed. Allegro does not expressly disclose at least one predetermined HMM or each of the plurality of predetermined HMMs comprises between 2 to 10 states. However, Rahim teaches a set of HMMs each having 3 to 4 states (See col. 14, lines 4-10). It would have been obvious to one of ordinary skill in the art at the time of the invention to limit the number of states to less than 10 to allow faster processing and reduced memory requirements.

Regarding claim 15, Allegro discloses a hearing aid prosthesis (1) comprising: a microphone (2) adapted to generate an input signal in response to receiving an acoustic signal from a listening environment, an output transducer (6) for converting a processed output signal into an electrical or an acoustic signal, processing means (4,5,7) adapted to process the input signal in accordance with at least two predetermined signal processing algorithm and respective sets of algorithm parameters to generate the processed output signal, a memory area (8) storing values of the related algorithm parameters for the at least two predetermined signal processing

algorithms, and processing the characteristics of the input signal with at least one discrete Hidden Markov Model, $\lambda^{source} = \{A^{source}, B^{source}, \alpha_0^{source}\}$, associated with a predetermined sound source to determine element value(s) of a classification vector indicating a probability of the predetermined sound source being active in the listening environment, control a transition between the at least two predetermined signal processing algorithms in dependence of the element values of the classification vector; wherein; A_{source} = A state transition probability matrix; B_{source} = An observation symbol probability distribution matrix for an input observation O(t) for each state of the at least one Hidden Markov Model (HMM); α_0^{source} = An initial state probability distribution vector (See Fig. 1 and paragraphs 0010-0027). Allegro does not expressly disclose segmenting the input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, O(t), representing predetermined signal features of the consecutive frames, wherein the feature vectors comprise a plurality of frequency-domain parameters or a plurality of time-domain parameters or any combination thereof, compare each of the feature vectors, O(t), with a feature vector set to determine, for substantially each feature vector, an associated symbol value so as to generate an observation sequence of symbol values associated with the consecutive signal frames. However, Allegro teaches extraction of characteristic features from an acoustic signal during an extraction phase (See paragraph 0019). Further, Rahim teaches a speech recognition system using a feature extractor to generate a set of feature vectors and a classification processor to generate a set of recognition models to be processed by one or more HMMs (See col. 6, line 26 to col. 7, line 35). Since speech occurs in the time-domain, it is inherent that the feature vectors comprise a plurality of time-domain parameters. Frequency-domain conversion is well-known in the art and is frequently used for

efficient data processing. It would have been obvious to one of ordinary skill in the art at the time of the invention that input segmentation and vector generation is an integral part of processing an input signal using a Hidden Markov Model for determining the real time status of the acoustic scene.

Regarding claim 17, Allegro discloses a prosthesis as claimed. Allegro does not expressly disclose a predetermined sound is a natural or synthetic sound source selected from a group consisting of: {speech, telephone speech, traffic noise, multi-talker or babble noise, subway noise, transient noise, wind noise}. However, Rahim teaches a predetermined source including various forms of speech (See Col. 13, lines 19-58). It would have been obvious to one of ordinary skill in the art at the time of the invention that the predetermined source is speech since the device itself is a hearing device for providing clear sound inputs.

Regarding claim 22, Allegro discloses a hearing aid prosthesis (1) comprising: a microphone (2) adapted to generate an input signal in response to receiving an acoustic signal from a listening environment, an output transducer (6) for converting a processed output signal into an electrical or an acoustic signal, processing means (4,5,7) adapted to process the input signal in accordance with a predetermined signal processing algorithm and related algorithm parameters to generate the processed output signal, a memory area (8) storing values of the related algorithm parameters for the predetermined signal processing algorithm, the processing means being further adapted to: process the feature vectors with a set of Hidden Markov Models and a make the hearing prosthesis capable of voice activation of user input devices(See Fig. 1 and paragraphs 0010-0027). Allegro does not expressly disclose segmenting the input signal into consecutive signal frames of time duration, T_{frame}, and generate respective feature vectors, O(t),

representing predetermined signal features of the consecutive signal frames, wherein the feature vectors comprise a plurality of frequency-domain parameters or a plurality of time-domain parameters or any combination thereof, or modeling respective isolated words or commands to determine element values of a classification vector indicating a probability of an isolated word or command being spoken. However, Allegro teaches extraction of characteristic features from an acoustic signal during an extraction phase (See paragraph 0019). Further, Rahim teaches a speech recognition system using a feature extractor to generate a set of feature vectors and a classification processor to generate a set of recognition models to be processed by one or more HMMs and modeling respective isolated words or commands to determine element values of a classification vector indicating a probability of an isolated word or command being spoken (See col. 6, line 26 to col. 7, line 35). Since speech occurs in the time-domain, it is inherent that the feature vectors comprise a plurality of time-domain parameters. Frequency-domain conversion is well-known in the art and is frequently used for efficient data processing. It would have been obvious to one of ordinary skill in the art at the time of the invention that input segmentation and vector generation is an integral part of processing an input signal using a Hidden Markov Model for determining the real time status of the acoustic scene and using specific isolated words or commands for voice activation.

Regarding claim 23, Allegro further discloses the processing means are adapted to recognize voice commands from the user to control a voice-activated user input device (See paragraph 0027). Allegro does not expressly disclose controlling one or several functions of the hearing aid. However, it would have been obvious to one of ordinary skill in the art at the time of

the invention that the voice activation capabilities of the device be used to control the hearing prosthesis for ease of use by the wearer.

Regarding claim 24, Allegro discloses a hearing prosthesis as claimed. Allegro does not expressly disclose the HMMs used are left-right HMMs. However, the use of left-right HMMs is well known in the art and Rahim teaches a set of left to right HMMs (See col. 14, lines 4-10). It would have been obvious to one of ordinary skill in the art at the time of the invention to use left-right HMMs to more easily match the normal left-right pattern of human speech.

Regarding claim 25, Allegro discloses a hearing prosthesis as claimed above. Allegro does not expressly disclose a training of the set of Hidden Markov models has been performed on words or commands spoken by the user during a fitting session. However, Rahim teaches an off line training session consisting of utterances from multiple environments including varying age, sex, etc. (See col. 2, lines 8-13). It would have been obvious to one of ordinary skill in the art at the time of the invention that the train set is performed on words or commands spoken by the user for efficient classification during use.

Regarding claim 26, Allegro discloses a hearing prosthesis as claimed above. Allegro does not expressly disclose the processing means comprises a software programmable processor. However, Rahim teaches a processing means comprising a software programmable processor (See Col. 5, lines 22-44). It would have been obvious to one of ordinary skill in the art at the time of the invention to use a software programmable processor to save space and provided a smaller unit for a less visible prosthesis.

Regarding claim 27, Allegro discloses a hearing prosthesis as claimed above. Allegro does not expressly disclose a training of the set of Hidden Markov models has been performed

on words or commands spoken by the user during a fitting session. However, Rahim teaches an off line training session consisting of utterances from multiple environments including varying age, sex, etc. (See col. 2, lines 8-13). It would have been obvious to one of ordinary skill in the art at the time of the invention that the train set is performed on words or commands spoken by the user for efficient classification during use.

Allowable Subject Matter

Claims 6-9, 11-13, 16, 18, 19 and 21 are objected to as being dependent upon a rejected base claim, but would be allowable if rewritten in independent form including all of the limitations of the base claim and any intervening claims.

Response to Arguments

Applicant's arguments with respect to claims 1, 2, 5, 10, 14, 22-24 and 26 have been considered but are moot in view of the new ground(s) of rejection. A different interpretation of the previously applied reference has been applied.

Conclusion

Any inquiry concerning this communication or earlier communications from the examiner should be directed to Brian Ensey whose telephone number is 703-305-7363. The examiner can normally be reached on Mon-Fri: 8:00 - 4:30.

If attempts to reach the examiner by telephone are unsuccessful, the examiner's supervisor, Curtis Kuntz can be reached on 703-305-4708. The fax phone number for the organization where this application or proceeding is assigned is 703-872-9306.

Any response to this action should be mailed to:

Commissioner of Patents and Trademarks
Washington, D.C. 20231

Or faxed to:

(703) 872-9306, for formal communications intended for entry and for informal or draft communications, please label "PROPOSED" or "DRAFT".

Hand-delivered responses should be brought to Crystal Park II, 2121 Crystal Drive, Arlington, VA., Sixth Floor (Receptionist).

Information regarding the status of an application may be obtained from the Patent Application Information Retrieval (PAIR) system. Status information for published applications may be obtained from either Private PAIR or Public PAIR. Status information for unpublished applications is available through Private PAIR only. For more information about the PAIR system, see http://pair-direct.uspto.gov. Should you have questions on access to the Private PAIR system, contact the Electronic Business Center (EBC) at 866-217-9197 (toll-free).

BKE
June 18, 2004

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